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TELECONFERENCING TERMINALS

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The purpose of this study was to contribute to the development of facsimile teleconferencing standards that are consistent with the U. S. Government's requirements, and that meet the needs of the U. S. users. The two objectives are: investigate user and terminal requirements, and define the functionalities of the terminals, networks, and control management units; and based upon the results of the first objective, design protocols which address mutiple function control, automatic call setup, transfer of control between conferees, billing, network connectivity. These objectives are addressed in the following sections: "Type of Teleconference"; "User and Terminal Requirements"; "Teleconferencing Standardization Efforts"; "Multipoint Conferencing Capabilities"; "Recommendations"; and "Areas for Future Study".

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TELECONFERENCING TERMINALS

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FOREWORD

Among the responsibilities assigned to the Office of the Manager, National Communications System, is the management of the Federal Telecommunication Standards Program. Under this program, the NCS, with the assistance of the Federal Telecommunication Standards Committee identified, develops, and coordinates proposed Federal Standards which either contribute to the interoperability of functionally similar Federal telecommunication systems or to the achievement of a compatible and efficient interface between computer and telecommunication systems. In developing and coordinating these standards, a considerable amount of effort is expended in initiating and pursuing joint standards development efforts with appropriate technical committees of the International Organization for Standardization, and the International Telegraph and Telephone Consultative Committee of the International Telecommunication Union. This Technical Information Bulletin presents and overview of an effort which is contributing to the development of compatible Federal, national, and international standards in the area of facsimile. It has been prepared to inform interested Federal activities of the progress of these efforts. Any comments, inputs or statements of requirements which could assist in the advancement of this work are welcome and should be addressed to:

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TELECONFERENCING TERMINALS

October, 1991

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1.0 INTRODUCTION

This document summarizes work performed by Delta Information Systems, Inc. (DIS), for the National Communications System, Office of Technology and Standards. This office is responsible for the management of the Federal Telecommunications Standards Program, which develops telecommunications standards, whose use is mandatory for all Federal department agencies. The purpose of this study, performed under task order number 89-1 of contract number DCA100-89-C-0078, is to contribute to the development of facsimile teleconferencing standards that are consistent with the U.S. government's requirements, and that meet the needs of the U.S. users. There are two objectives:

- o Investigate user and terminal requirements, and define the functionalities of the terminals, networks, and central management units.
- o Based upon the results of the first objective, design protocols which address multiple function control, automatic call setup, transfer of control between conferees, billing, and network connectivity.

To address these objectives, this report is divided into six sections.

Section 2.0, "*Types of Teleconferencing*," discusses the different types of teleconferencing, and the parameters affecting them.

Section 3.0, "*User and Terminal Requirements*," discusses what users might want from a multimedia terminal, and its impact on teleconferencing.

Section 4.0, "*Teleconferencing Standardization Efforts*," discusses efforts by the CCITT to standardize teleconferencing.

Section 5.0, "*Multipoint Conferencing Capabilities*," discusses network connectivity, automatic call setup, transfer of control between conferees, multiple function control, and billing.

Section 6.0, "*Recommendations*," recommends how multipoint conferencing capabilities might be standardized.

Section 7.0, "*Areas for Future Study*," suggests areas of future study to better define teleconferencing.

2.0 TYPES OF TELECONFERENCING

In general, there are three types of teleconferences: audio, audiographic, and video. None is superior to the others; each fulfills a specific task, and each will be explored in the following sections.

2.1 Audio Conferences

Audio conferencing allows two or more people at two or more locations to communicate, usually via the Public Switched Telephone Network (PSTN). All participants can hear and be heard by one another, but no use is made of the participants' visual sense unless visual materials were circulated in advance.

With the PSTN as its basis, audio conferencing is, at present, one of the least limiting teleconferencing technologies. The PSTN allows people in even the remotest of outposts to participate. Although participants can use a regular telephone handset, speakerphones and bridging networks allow several people at one location to hear and talk to participants at other locations.

Thousands of audio conferences take place in the United States every day, and conference participants are connected by either phone company conference operators, commercial bridging services, or private audio bridges. According to the companies offering teleconference services, the average audio conference links seven or eight locations[1] for either business or training meetings. Business meetings usually average 45 minutes, with one person at each location, and training conferences last 90 minutes with more than one person at each location. These are averages. Audio conferencing can (and does) work for 15-minute updates among three people or for day-long meetings of thousands.

Although audio teleconferencing is considered the most basic (and least sophisticated) means of teleconferencing, it is critical to the successful operation of the "more sophisticated" teleconferencing methods (audiographic and video). That's because most communications between humans is through the spoken word. Without audio, the sophisticated teleconferencing methods lose their effectiveness. In fact, 57% of today's teleconferences use interactive audio only (See Table 2-1).[2],[3]

Having audio, however, does not automatically ensure a successful teleconference. The *quality* of the audio must allow users to hear one another clearly, and the teleconference itself must be well managed. What factors influence sound quality, and how teleconferences can be better managed are both discussed.

Table 2-1. Teleconferencing Use Breakdown

Type of System	Percent of Total
Audio teleconferencing	57.14
Meet-me bridging	14.97
Dial-up conferencing	22.45
Dedicated network	12.93
Meet-me and dedicated	2.72
Dial-up and dedicated	3.40
Audio and graphics	11.56
Facsimile	5.44
Electronic blackboard	2.72
Electronic blackboard and facsimile	0.68
Electronic tablet and facsimile	0.68
Typewriter or computer	2.04
Audio and freeze-frame/slow-scan video	8.16
Audio, freeze-frame and graphics	7.48
With facsimile	4.76
With electronic blackboard	0.68
With electronic blackboard and facsimile	1.36
With electronic blackboard, facsimile and computer	0.68
Audio and full-motion videoconferencing	12.24
Audio, full-motion video and other	3.40
With facsimile	0.68
With freeze-frame	1.36
With freeze-frame and facsimile	0.68
With freeze-frame and computer	0.68

2.1.1 Sound Quality

A clear and intelligible audio link is crucial to the success of an audio conference. If participants have difficulty hearing one another then their teleconference will probably be a

failure. The clearness and intelligibility of the audio link depends on at least three items: the transmission line, the conference rooms' acoustics, and the terminal equipment (microphones and speakers).[4],[5],[6] The last two are important because participants hear what the microphones hear. So, if conference room acoustics are poor, or if the microphones work poorly, the sounds transmitted by the microphones to conference participants are likely to be unintelligible.

2.1.1.1 Room Acoustics

As mentioned before, a room's acoustical properties play an important role in shaping the sounds heard by microphones. These properties can be broken down into three main categories: echoes, reverberation, and background (ambient) noise. The first two describe a room's reaction to sounds; while the last one describes sounds which are always present in the room. Echoes are exact repetitions of original sounds. A good example is the sound of your own voice returning in a large cavern. Reverberations are multiple echoes with very short times in between. They are perceived as sounds which "bounce around" and are generally heard as a hollowness in voices. Excessive reverberation can make it difficult for listeners to understand a conference speaker and can make a room unsuitable for audio conferencing. Background or "ambient" noises are sounds a room contains even when unoccupied, and they have many sources: air-conditioning, heating and ventilation equipment, office equipment, lighting fixtures, outside noises, etc.

Reducing a conference room's echoes, reverberations, and ambient noises, may require modifying the room. For instance, sound absorbent material placed on room surfaces, or multi-angled walls and ceiling can limit the reflection of sound. Nevertheless, their application must be done judiciously and must account for all sound frequencies. For example, a common mistake is to assume that general purpose acoustic absorption materials, such as carpeting, acoustic tiles, and thin fiberglass panels, will absorb lower frequency sounds when placed directly on'a surface. They might not, and, unfortunately, a "boomy" or "hollow" sounding room can result.

2.1.1.2 Transmission Lines

Electrical noise and distortion, associated with the transmission line can also hamper an audio conference. The electrical noise can come from amplifiers, power line induction, crosstalk, signal frequency tones, etc. Since, most sounds are, at present, carried over transmission lines encoded in analog form - a continuously varying electromagnetic waveform, when analog transmissions are amplified any noise is also amplified. Thus it's possible, in analog transmissions, for electrical noise to drown out the desired sounds.

Fortunately, telephone companies, the most used media for teleconferences, are going "digital". Digital transmissions naturally separate desired sounds from unwanted electrical noise. With digital, sounds are modeled as a time series of discrete numbers expressed in binary form. Since just ones and zeros are transmitted, noise is not amplified in digital transmissions. Other advantages are that computers, since they share the same language, can process digital transmissions economically, compress the signals to achieve better bandwidth, and, if necessary for security reasons, encrypt the signals.

To take advantage of these features, the telephone companies are making the PSTN into an error-free digital network by converting their copper-based systems to fiber-optic-based systems. These all-digital systems will eventually provide the user with a number of combined services which include voice, facsimile, data, and video, and will be known as the Integrated Services Digital Network (ISDN).[7]

2.1.1.2.1 ISDN

ISDN is not yet commonplace, nor are all its possible uses understood or defined. A major hurdle is the large number of copper wires connecting users to local exchanges. While telephone companies convert the established copper base to fiber-optics, they want all their users to be able to take advantage of ISDN, and they think they have found a way. They have determined that they can transmit 144 kb/s to each user over existing copper wires. This service, called the Basic Access Service, provides the user with two 64 kb/s (B) channels and one 16 kb/s (D) channel. The Telephone companies demonstrate the use of these channels with the example of two phone users exchanging data during a phone conversation. Nevertheless,

all the possible uses for these channels are as yet unknown. For instance, the channels might be used for the following purposes:

- o To transfer data at 144 kb/s
- o To multiplex and connect a number of computer users to a remote computer
- o To conduct an audio conference during which facsimiles of conference materials are transmitted to all participants.

2.1.1.2.2 Transmission Line Operation

How the transmission line operates also has an effect on the clarity and intelligibility of a teleconference. For instance, the most natural sounding audio gives users the ability to talk and listen at the same time. Providing this capacity for simultaneous two-way conversation can be accomplished by using a "full-duplex" channel. Full-duplex uses two channels or paths where one channel allows a user to listen, and the second channel allows the user to speak, both simultaneously. Messages are sent and received in both directions at the same time. Secondly, because the channels are separated, unwanted feedback signals are eliminated.

Another method uses just one channel for both talking and listening (half-duplex); unfortunately, it is less natural sounding. With half-duplex a message may only be sent and received in one direction at a time. To accommodate messages from more than one speaker, the channel must switch from one message to another. Typically, switching is based upon sound levels. So, when users converse on a half-duplex transmission line, it is usually the loudest whose voice is heard.

At present, a regular dial-up phone is often all that is needed to conduct an audio conference between two locations. When more than two locations are involved, more complex equipments (audio bridges) are needed to allow the participants to interact. Audio bridges connect all participants and control the audio levels. For half-duplex transmission lines, audio bridges can supply two types of switches: a hard or soft switch. A hard switch opens in response to a message and stays open until that message completes. During the message, no other speaker can interrupt unless there is a pause that is long enough for the switch to close. Since a hard switch takes a few milliseconds to respond, it usually creates some unevenness in the flow of conversation, especially during rapid exchanges. Hard switches can be frustrating

to users who are unable to interrupt the speaker or who are blocked out. Unlike a hard switch, a soft switch allows more than one speaker to be heard at the same time. The switch provides a balance between the primary speaker and secondary speakers. The primary speaker is transmitted at full audio, while secondary speakers are muted. With a soft switch, the primary speaker can be interrupted and switching from one speaker to another is not as abrupt.

2.1.1.3 Terminal Equipment

In addition to telephone hand-sets and speakerphones (hands-free amplified telephones) there are several options for speaker-microphone equipment involving both their type and how they are operated. The most frequently used microphones are omnidirectional microphones placed or mounted on conference tables, and lavaliere microphones which are hung around necks. In large conference settings, highly directional shotgun microphones are invaluable. They avoid forcing participants to line up at microphones and eliminate pauses while participants make their way to a microphone. The microphones can be used in three ways: voice-switched, press-to-talk, and open audio.

Voice-switched eliminates sounds transmitted from all microphones other than the one into which a participant is speaking. This reduces ambient noise and prevents sound from a loudspeaker from reentering the transmission circuit. When another participant speaks louder than the incoming sound, their microphone is automatically turned on. However, loud ambient noise or other noises, such as a cough, can activate the microphone and keep it on, blocking all other locations. Voice-switched microphones need a relatively quiet environment to operate properly. Also, since voice-switched microphones allow only one person to talk at a time, it can be difficult to interrupt a speaker. Some voice-switched microphones have a softer switching mechanism which mutes but does not completely block simultaneous messages; thereby accommodating the more natural flow of human conversation.

Press-to-talk microphones are not active until a user depresses a button. This keeps ambient noise out of the audio link when a participant is listening and it cannot be accidently activated by loud noises in the room. Press-to-talk microphones are very useful in noisier room environments and on larger teleconferencing networks where noise is compounded by the number of on-line sites.

Open audio microphones transmit sound continuously. They are usually omnidirectional and pick up sound from all directions in a room. Therefore, it is crucial that excess noise be eliminated. Nevertheless, these microphones are preferred because people find them easy and natural to use.

2.1.2 Conference Management

A number of studies have shown that planning and effective management are key to a successful teleconference.[8],[9] Without them a teleconference risks failure. A good teleconference includes the following:[10]

1. It is well-planned and organized, with a clear picture of how the teleconference will accomplish meeting objectives.
2. The chairperson is familiar with the teleconference equipment, especially the proper use of microphones, and knows what to do if technical problems arise.
3. The chairperson plays a strong role in managing the meeting: focusing on the agenda, controlling discussion, and establishing protocols for proper meeting behavior, such as speaking order and speaker identification.
4. To overcome geographical distance, humanizing techniques are used to create group rapport and to acknowledge individual participants.
5. Techniques are used to control underparticipation or overparticipation.
6. Messages are presented clearly so that they are received, understood, and remembered.
7. Feedback is elicited from participants to correct misunderstandings, fill in omissions, and plan future meetings.

2.2 Audiographic Conferences

When the exchange of spoken words is not enough, an audio conference can be enhanced with electronic graphics which can display or print charts, diagrams, text materials, and still video images of people or objects. Audiographic conferences allow participants to talk easily to one another and simultaneously view and interact with the same visual materials.

A large number of devices can be used to turn an audio conference into an audiographic conference. Some of these devices are: electronic pens, blackboards, and tablets, computer systems, facsimile machines, and slow-scan (freeze-frame) video (See Table 2-2).

Table 2-2. Graphic Options

Device	Capability	Examples
Facsimile	Paper reproductions, Delayed transmission, Permanent paper copies	Typewritten pages, Documents, Prepared graphics, Pictures
Telewriters	Hand-drawn graphics, Shown on TV monitors/ projection screens Instant transmission	Writing, Drawings, Outlines, Equations, Graphs
Computer Systems	Text and computer-drawn graphics, Instant transmission, Shown on TV monitors, Permanent paper copies	Alphanumerics, Diagrams, Graphs, Schematics, Charts
Random access microfiche and slide projectors	Microfiche images Slides	
Slow-scan television	TV Resolution	

Electronic pens, blackboards, and tablets are usually grouped together as telewriting systems, because they allow hand-drawn information to be generated and sent to remote locations. Whereas, facsimile systems transfer paper documents over distance.

2.3 Video Conferences

Of all the teleconferencing technologies, the video conference most closely approximates an in-person meeting. It allows participants to have a more intimate and active exchange of visual information. Today, the video conference is primarily a point-to-point technology. Corporations are the main users of video conferencing and are used primarily for project management, intergroup coordination, management meetings, and information dissemination.

3.0 USER AND TERMINAL REQUIREMENTS

In the past, teleconferencing systems were custom-built to each customer's needs. Unfortunately, this usually made different customers' systems incompatible with one another. This section explores a theoretical multimedia system to determine what users want, and thereby define a set of possible user-related terminal requirements.

3.1 User Requirements

To date, teleconferencing systems have usually been customized to a particular customer's needs. Thus their system will work fine within it's own world, but may not work when connected to someone else's system. To help define the recommendations specifying systems' interoperation, the needs of the user should be examined.

One way to do this is to survey users and potential users of teleconferencing systems, and allow them the pick the desired services. Robert Johansen, in his book "*Teleconferencing and Beyond: Communications in the Office of the Future*" describes such a survey.

Survey participants were allowed to pick multimedia services which best suited their needs. The system could consist of inputs, outputs, and computer-based functions (See Table 3-1). In the table, the meaning of most of the inputs and outputs are self-explanatory. A few are not. For instance, page input means to scan a document, while video input can be still or motion video. Also, page output is a facsimile-style output, while typed page output is defined as high-quality *original* documents. Finally, still video output includes CRT displays.

Table 3-1. Hypothetical System Services

Inputs	Outputs
Audio	Audio
Keyboard	Typed Page
Page	Page
Graphics	Still Video
Video	Motion Video

Computer-Based Functions
Graphics
Numerical Processing
Text Processing
Database

In the survey, participants selected multimedia services by connecting inputs to outputs. They could connect as many inputs to as many outputs as they wished. The results suggest the following specific pairings:

70% - audio-to-audio (audio teleconferencing or voice mail)
70% - keyboard-to-typed page (Typewriter)
56% - page-to-page copy (Hard-copy Facsimile)
52% - keyboard-to-still video (CRT)
52% - audio-to-typed page (Talking typewriter)
44% - graphics-to-still video (Soft-copy electronic tablet)
39% - video-to-motion video (Video conference)
35% - page-to-typed page (Character Recognition)
32% - page-to-still video (Soft-copy Facsimile)
32% - graphics-to-page copy (Hard-copy electronic tablet)
32% - keyboard-to-page copy
31% - graphics-to-typed page

To determine the services participants felt they really needed, they were asked to select just four multimedia services (also considered a low-budget system). The results stress those services participants felt were basic multimedia components:

66% - keyboard-to-typed page (Typewriter)
58% - audio-to-audio (audio teleconferencing or voice mail)
38% - keyboard-to-still video (CRT)
36% - page-to-page copy (Hard-copy Facsimile)
33% - audio-to-typed page (Talking typewriter)

In both systems, the most desirable *teleconferencing* services were audio and hard-copy facsimile. Soft-copy facsimile, telewriting, etc., took a back seat.

3.2 Terminal Requirements

Based upon the results of section 3.1, "*User Requirements*," a teleconferencing terminal should, at a minimum, provide audio service. All other services, facsimile, graphics (electronic tablets/blackboards, etc.), electronic mail, video, etc., should be optional with hard-copy facsimile receiving preference. With this in mind, it is clear that different terminals could have different service capabilities when they are communicating with one another. For instance, in a two-terminal connection, one terminal might provide just audio; while the other terminal might

provide audio and video. Since the audio only terminal is unable to provide video to the other terminal, and since a similar scenario would occur between other dissimilar terminals, it would seem reasonable to stipulate that communications between dissimilar terminals may only use services common to both. In the aforementioned example, audio would be the common service. Nevertheless, before the terminals allow any services to be used, they should negotiate which services can be provided to their users. So, in a multi-terminal conference where there are many terminals with different service capabilities, to provide a user with a particular service, at least two of the terminals must have that service capability.

In addition to negotiating service capabilities, each service might have their own levels of capability, and those levels in turn might have their own levels of capability, and so forth. For instance, for facsimile, there are Group 3 and Group 4 compatible equipments, and within Group 3 and Group 4, a number of items must be negotiated, such as pixel resolutions, page size, etc. So, a teleconference employing facsimile would have to determine which terminals have facsimile capability, negotiate Group 3 or Group 4 compatibility, and then negotiate pixel resolution, page size, etc., before using the facsimile service.

4.0 TELECONFERENCING STANDARDIZATION EFFORTS

Today, there are many companies making teleconferencing equipment. Unfortunately, equipment made by one company may be unable to interoperate with equipment made by another company. To get different companies' equipment to interoperate, at least one international standardization group is developing teleconferencing standards. The International Telegraph and Telephone Consultative Committee (CCITT) is developing standards, known as Recommendations, which describe the general aspects of teleconferencing, and which define the requirements needed to allow different companies' equipment to interoperate. Their efforts have yielded a set of Recommendations which describe what audiographics teleconferencing is, and what is needed to achieve interoperating equipments. At present, they are working on a number of Recommendations which will describe how the equipments will interoperate. The following sections discuss portions of the teleconferencing Recommendations already developed, and discuss the issues pertaining to those under development.

4.1 CCITT's Audiographic Conference Service

The CCITT's draft Recommendation pertaining to an international audiographic conference service defines a teleconferencing model, and, based upon the model, divides the teleconferencing service into two parts: basic service and optional service.[11] In addition, the Recommendation discusses operational procedures, quality of service, and terminal and network requirements.

4.1.1 Functional Teleconferencing Model

The functional model provides the base for the basic and optional teleconferencing services by defining the roles of conference members and the implied exchange of information in an audio conference.

4.1.1.1 Roles of Conference Members

In the functional model, conference members can assume one or more of four roles:

1) Conference convener
2) Conference conductor (or controller)
3) Presenter
4) Audience member

The first two, as their names imply, are responsible for convening and conducting (controlling) the conference. A *conference convener* (See Figure 4-1) arranges and reserves the conference facilities, and summons conference participants; the *conference conductor* sets up, manages (chairs), and clears the conference. Since managing the conference may also require controlling who has the floor, the conference conductor may have to coordinate and manage network and terminal functions during the conference. In addition, since the conference conductor may be unfamiliar with the technical aspects of teleconferencing, coordinating and managing network and terminal functions should require no special training.

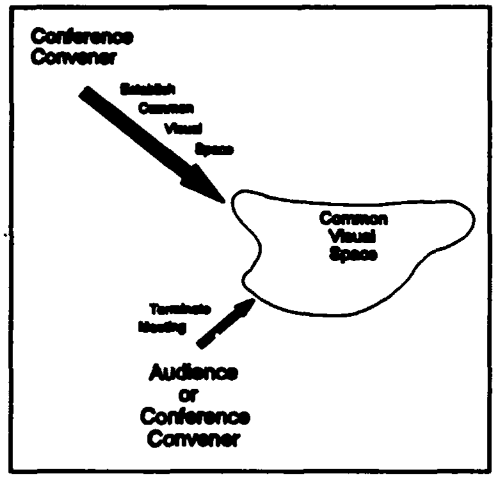

Figure 4-1. Establish/Terminate Conference Session

The last two, the presenter and the audience member have no explicit control over the conference itself. Nevertheless, it is for their benefit that conferences are held; conferences allow them to exchange information (See Figure 4-2). Specifically, the conference presenter presents information to the audience for the audience's benefit, and the audience listens and learns (hopefully) from the information presented. If a member of the audience wants to comment on the information presented, or wants to become the conference presenter, he or she does so by following the

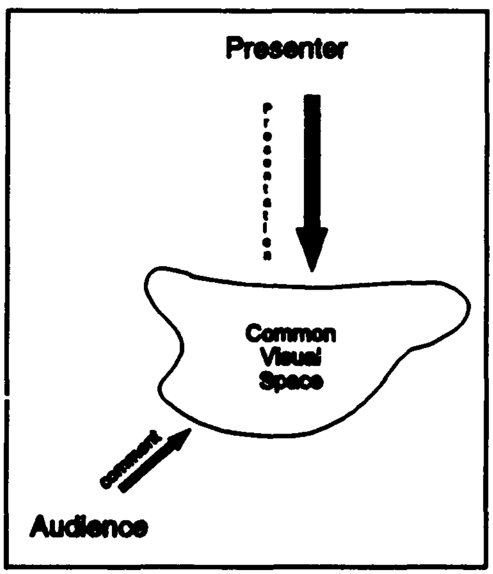

Figure 4-2. Active Conference

rules established by the conference controller. The rules are usually simple and can range from no rules at all to a required formal acknowledgment by the conference conductor to the request (See Figure 4-3). Which method is used depends on the size of the conference. For small conferences, formal acknowledgement is usually unnecessary: all participants can easily converse with one another. For large conferences, formal acknowledgement is almost a necessity: a large conference can become a disaster if potential conference presenters must joust for speaking privileges.

Of these four roles, a single individual can perform all but two simultaneously, in a conceptual sense. The two exceptions are conference presenter and audience member. For instance, a single individual might act as the conference convener, the conference conductor, and the *conference presenter*, or as the conference convener, the conference conductor, and an *audience member*, but a single individual can not be both the conference presenter and audience member simultaneously. The roles are opposites, one must be traded for the other (These conceptually diametric roles should not to be confused with technology's ability to treat the conference presenter as though he too were an audience member).

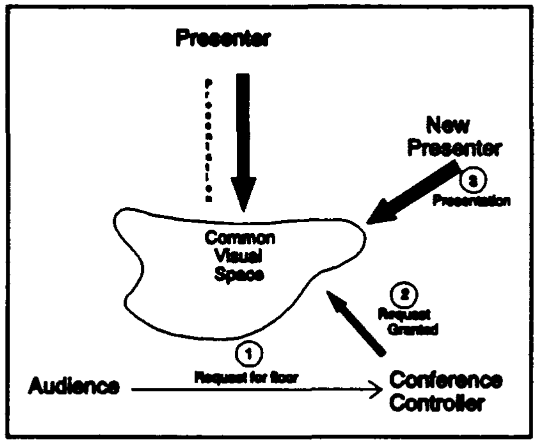

Figure 4-3. Presenter Interchange

4.1.1.2 Implied Exchange of Information

During a conference, two types of information are exchanged: audio and graphics (imagery) information, and conference control information (establishment, disestablishment, etc.). Audio and graphics information can be exchanged by all four role models. Whereas conference control may only be handled by the conference convener and conference conductor.

4.1.2 Basic Service Requirements

The basic service requirements specify the minimum level of interoperability between audiographic terminals, and specify allowed terminal and conference capabilities. For instance,

according to the requirements audiographic terminals *may* provide audio, imagery, and annotation; but, only audio *must* be provided. Nevertheless, a minimum level of interoperability between terminals for each capability must exist, if that capability is to be taken advantage of. Other requirements are as follows:

1) All terminals must provide audio without interruption.
2) Received imagery may be printed or recorded according to the wishes of the recipient and the capabilities of the terminal.
3) The service can use any type of network.
4) The service allows private use applications, such as encryption.
5) Multipoint conferences are controllable from a single location.
6) No fundamental changes are made to Recommendations for existing services or networks.

4.1.3 Optional Service Requirements

Optional service requirements extend the basic requirements to allow more sophisticated terminals to interact with one another and with less sophisticated terminals. For instance, some optional services are auto-call setup, editing, local view, etc. In general, the optional services must allow for the following:

1) Different pel transmission densities
2) Optional coding schemes
3) Grey scale images
4) Color images
5) Printable areas
6) Escape into national and private options
7) Resolution conversion algorithms
8) Confidentiality or security
9) Document control and editing functions
10) Audio requirements (enhanced quality, etc.)
11) Active participant indicators (speaker, presenter, etc.)

4.1.4 Operational Procedures

The operational procedures are based upon the major sequence of events which occur during an audiographic conference (See Table 4-1). These events can be broken into three phases (See Figure 4-4):

time -------->

Phase A1	Phase B1	Phase C	Phase B2	Phase A2

Figure 4-4. Phases of an Audiographics Conference

1) Call setup and disconnect (Phases A1 and A2)
2) Conference setup, recovery, and reconfiguration (Phases B1 and B2)
3) Conference session (Phase C)

4.1.4.1 Call Setup and Disconnect

The establishment and disconnection of the physical connection between terminal equipment for point-to-point operation, or between terminal and MCU for multipoint operation.

4.1.4.2 Conference Setup, Recovery, and Reconfiguration

As soon as a physical connection exists, the terminal enters the conference setup phase. During the setup phase, participating terminals exchange terminal identity and authentication, names of participants, and terminal characteristics and capabilities. In addition, if during the conference, participants change, then the setup phase can be reentered.

During conference recovery or reconfiguration, the terminal controlling the conference is responsible for detecting if a connection to any terminal is broken. If the controlling terminal is lost, conference setup procedures are re-initiated.

Table 4-1. Sequence of Events During an Audiographics Conference

1) Reservation - Facilities are reserved for the conference for a particular time and date.
2) Establish communications - At the time and date agreed upon, audio and imagery communications are established between all terminals participating in the conference.
3) Originate an image - The presenter transmits an image that is simultaneously displayed at all participating conference terminals.
4) Work with image - The presenter may activate an electronic pointer or annotate the image; all locations immediately see the results of this action (Real-time operations).
5) Modify the image - Locally edit the image; when transmitted, it will replace the current image at all terminals.
6) Record the image - Any audience member can record a copy of the currently displayed image on hardcopy or electronic media at any time without disrupting the conference.
7) Continue the conference - Conference participants repeat steps 3 through 6 with any participating terminal capable of representing a presenter and able annotate the work of others.
8) Terminate the conference - End the conference.

4.1.4.3 Conference Session

Two modes of operation may be used during a conference: non-conducted mode and conducted mode.

During a non-conducted mode session any participant may speak or initiate graphics activity at any time. This is the default mode at the start of a conference.

During a conducted mode session a participant must request to speak or perform graphics activity, and wait for permission to be granted. Permission is granted by the conference conductor, and can be granted manually or verbally, or by a queuing mechanism. In addition, the role of conference conductor can be transferred to another terminal at any time during the conference as long as that terminal can handle the role.

For either case, the conductor retains the capability to interrupt the current presenter, to disable the current presenter's microphone, and to give the floor to a different presenter.

4.1.5 Quality of Service

Quality of service for audiographics refers primarily to voice and graphics quality. By design of the conference rooms or the equipment or both, participants should be able to listen

and speak simultaneously with the voice quality achievable with speech transmission using a bandwidth of 7 Khz. Only if the transmission of graphical information competes for this bandwidth may this requirement be relaxed, and only if speech quality is not seriously degraded. For graphics, transmissions and image manipulations should be accomplished within the nominal response time associated with the related face-to-face activity.

4.1.6 Network Requirements

The draft Recommendation specifies that the audiographics service must be able to provide point-to-point and multipoint configurations with full interactivity among all terminals in the conference, and also specifies that a broadcast mode should be available.

Multipoint operation is provided by one or more multipoint control units (MCUs): where MCUs may be interconnected, and each MCU may serve one or more audiographic terminals.

At present, operator assistance is necessary to establish conference calls on the PSTN. Hopefully, this may not be necessary in the future, and is addressed in the Supplements to the CCITT Series E Recommendations.

Lastly, the CCITT intends to optimize the service for ISDN operation with integrated audio and graphics. Nevertheless, they would still like it to be compatible with the PSTN, the Circuit Switched Data Network (CSDN), and the Packet Switched Data Network (PSDN) where audio and graphics may be carried in separate circuits.

4.1.7 Terminal Requirements

According to the CCITT in its draft Recommendation, an audiographic terminal (AGT) is meant to imply all equipment used to input, output, and process both audio and graphics as associated with the audiographic service. Some examples are scanners, cameras, electronic tablets and keyboards, computer systems, printers, video displays, facsimile equipments, conference control equipment, microphones, speakers, etc. Depending on user needs, this equipment may be in one device or separate devices.

In addition, an AGT should allow simultaneous viewing, manipulation, and oral discussion of the same image on multiple AGTs linked by one or more networks. Although both hard and soft copy imagery can be transmitted and discussed, the intent is to use soft copy imagery as the primary conveyor of graphic information. This allows all participants to view documents on AGT displays, and use voice, pointers, and annotation devices during discussions about the documents.

4.2 Issues Being Resolved

The CCITT Study Group VIII/Q23 is developing teleconferencing Recommendations and works on several levels simultaneously. For instance, while defining high-level aspects, such as what the service entails, they might also be working on low-level aspects, such as the format of transmitted data. Since, in general, what the service is has been defined, the Study Group is now concentrating on how the service can be provided. Their efforts are following several avenues. For instance, they have already decided how teleconferencing data will be carried over various communication networks (CCITT Recommendation H.221, etc.), and they have decided that the higher levels of the teleconferencing protocols will consist of two parts: a Conference Control Protocol/Service (CCS), and a Multipoint Communication Protocol/Service (MCS)). Furthermore, they are zeroing in on what layers of Open Systems Interconnection (OSI) will be used to support the CCS and the MCS, and are deciding how various teleconferencing functions might be divided between the CCS and the MCS.

4.2.1 OSI Adherence

OSI consists of a seven-layer model or framework which ensures that all new communication standards are compatible. OSI is being defined by the International Organization for Standardization (ISO) whose primary goal is to define standards to allow different systems to communicate, with a secondary goal of retaining existing standards whenever possible.[12],[13] A system obeying the OSI model in its communication with other systems is termed an "open system". The OSI open systems concept allows application processes to interact with any other application process anywhere in the world.

The seven layers of the OSI model are divided among three different functions: user interaction, interface, and communication network interaction (See Figure 4-5).

Layer	Function
7 Application 6 Presentation 5 Session	User Interaction
4 Transport	Interface
3 Network 2 Data Link 1 Physical	Communication Network Interaction

PSTN
ISDN
PSDN
LAN

Figure 4-5. The OSI Model

The seven layers have the following definitions:

Application	-	The highest level. It is the user interface between a service and the OSI environment.
Presentation	-	The presentation layer handles session establishment and termination requests, and it preserves the meaning of data while resolving syntax differences.
Session	-	The session layer establishes, manages, and releases the communication connection.
Transport	-	Acts as a consistent interface between the application-related functions and the transmission-related functions.
Network	-	Provides routing and relaying through switched telecommunication media.
Data Link	-	Reliably transfers all information over the physical transmission media.
Physical	-	Deals with the transmission of a bit stream, regardless of its meaning, across a physical communication medium.

The CCITT Teleconferencing Study Group is concentrating on what layers of OSI will be used to support the CCS and MCS functions. Five different proposals are being considered (See Figure 4-6). Under normal circumstances a fully implemented OSI structure might have been implemented without much discussion. Unfortunately, a fully implemented OSI structure may be unable to support real-time teleconferencing applications. For example, if a presenter were annotating a document during his or her's presentation, the timing of the annotations received on participants' AGTs might not coincide with the presenter's speech. Such a situation could cause confusion or misunderstanding about the point the presenter trying to make. So, some of the proposals suggest ways the overhead of OSI could be reduced to eliminate loss of synchronization. For MCS, the favored protocol stack is shown in Figure 4-7.[14]

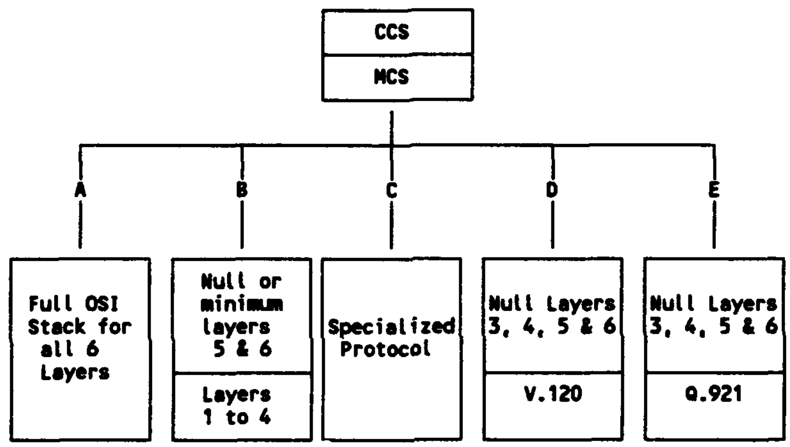

Figure 4-6. OSI Implementation in Teleconferencing

The roles of the CCS and the MCS have not yet been fully defined. Nevertheless, the CCS will probably provide all control aspects of a teleconference: i.e., establish or de-establish the conference, control who joins the conference, control who is the presenter, control network resources, etc., and the MCS will probably provide the underlying mechanism to accomplish the CCS functions. For example, the CCS might specify that a particular AGT will be transmitting a facsimile while all other able AGTs must receive the facsimile. The MCS would then negotiate common facsimile capabilities among the transmitting and receiving AGTs. Other issues being considered are aggregate acknowledgments, tokens, and active conference information.

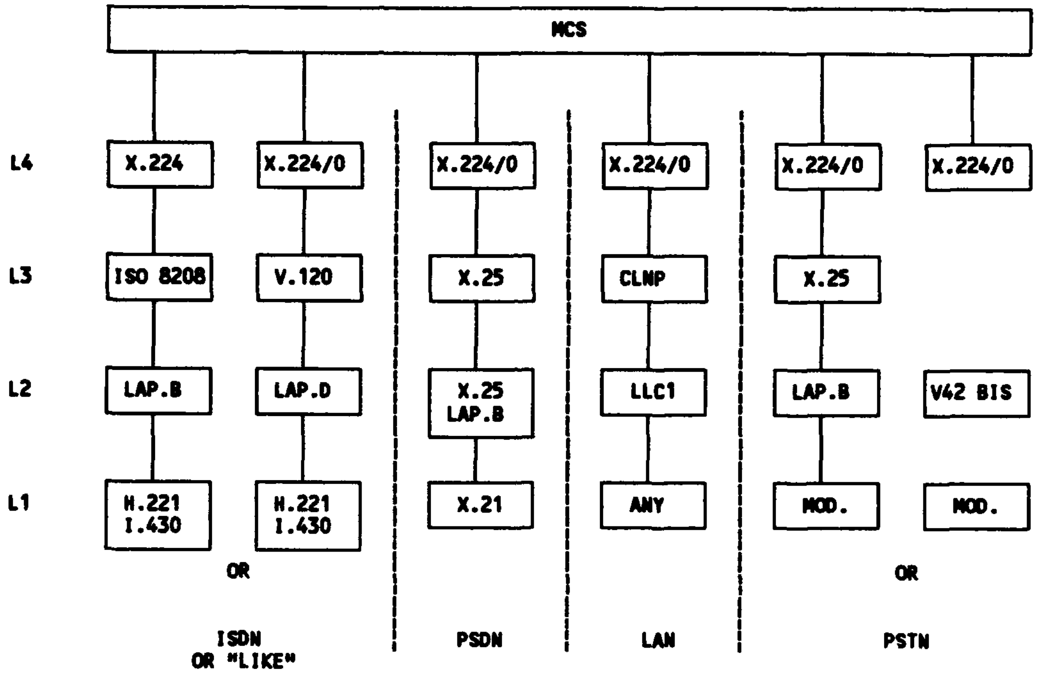

Figure 4-7. Protocol Stacks by Network

4.2.2 Multipoint Communications Service (MCS)

The MCS is in draft form and is defined as a generic, highly interactive, multimedia conferencing service capable of supporting full-duplex, multipoint communications between an arbitrary number of sites over a variety of networks.[15] It offers broadcast transmissions coupled with optional delivery confirmations. Furthermore, messages may be sent to all conference participants, subgroups of participants, or individual participants. Other services include token management and data synchronization.

4.2.2.1 Addressing

In MCS, users (clients) are usually addressless. Instead, clients rely on specific communication channels to transmit and receive information. These channels come in three flavors: reserved, multicast, and single-member channels. Reserved channels (empty, full, and controller) provide conferencing control and monitoring. For instance, the empty channel is restricted for use between a client and a MCU and may be used for token operations. Multicast channels provide communication between all or a subset of all clients in a conference. And,

single-member channels are for point-to-point communications or token operations. Together, these channels provide to all clients broadcast, limited broadcast, and point-to-point communications.

Clients receive messages by joining specific multichannels of interest, and who may join is controlled by the channel's "owner" (usually other clients), if they exist. Sending messages (on multicast channels) is done independently of channel membership: a client merely specifies the receiving channel without worry about whether or not he is a member of that channel. If a client wants to communicate privately with other clients, he (and they) may request a "user ID" which, if granted, is allocated as a single-member channel. With it, (and if both clients have it) clients may communicate on a point-to-point basis. Channel membership is recorded on a distributed basis at each level of a multiple-MCU domain.

4.2.2.2 Data Transfers

Data transfers typically assume multipoint delivery with an optional delivery confirmation. In general, the messages sent by the MCS are expected to be self-contained; i.e., no message reconstruction is expected to be done by the MCS. This is primarily for three reasons: 1) the message might be real-time (pointer control, for example), 2) other messages should not be blocked because of message reconstruction, and 3) it could be difficult to identify real-time messages and give them priority over message reconstructions. This doesn't mean extended messages are prohibited from being broken apart and then reconstructed; overlaid applications <u>can</u> do message reconstruction and are probably better suited to identify real-time messages.

Messages received by different clients are unlikely to be received in the same order. This is mainly because messages are expected to be sent using the most direct'route. Nevertheless, for those occasions where messages must be received in order, a "synchronized" send may be used. With it, all messages are sent to the main MCU which then dispatches them, in the same order, to each client, including the original sender.

Additionally, if a client wishes delivery confirmations, messages may be sent with a request for acknowledgement (At present, none is required). Since a message normally goes to more than one recipient the returning acknowledgements constitute a "many to one" transmission

and are condensed (at least for now) into one aggregate acknowledgement. For example, if the convener's AGT requests with acknowledgement all participants' AGTs to be connected to the conference, the convener's AGT would receive a single aggregate acknowledgement as opposed to individual acknowledgements from each participating AGT (See Figure 4-8). The format of the aggregate acknowledgement is still being debated with the currently proposed format appearing in Figure 4-9.

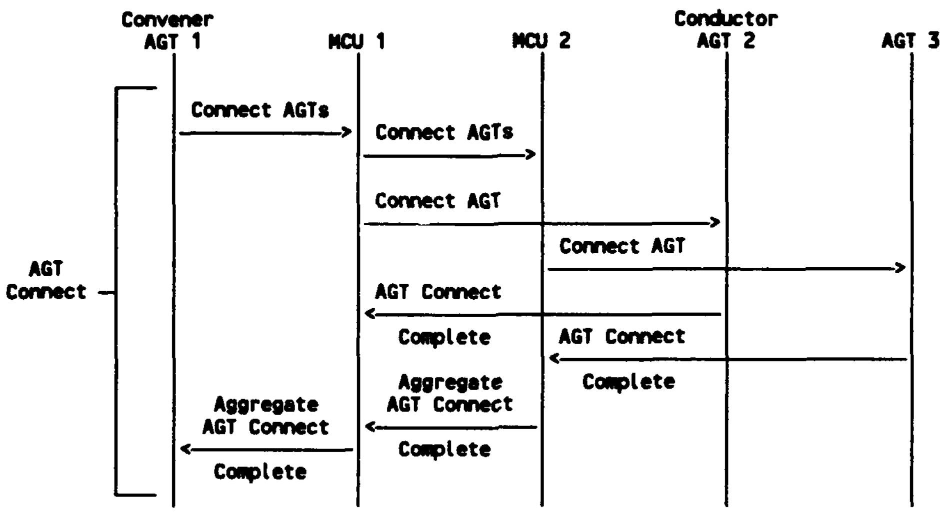

Figure 4-8. Example of Aggregate Acknowledgement

4.2.2.3 Tokens

Tokens provide the means to resolve resource contentions and to carry out exclusive events in a real-time environment. For example, in a multipoint environment employing pointers, one and only one client should control the pointer at any given moment. If a client wants to use a pointer, he must request the token; which will probably be granted if no one else is holding it. Tokens may be owed by clients who may also control who uses them. Actual control is granted by mapping tokens into reserved, single-member channels and temporarily making the client an exclusive member of that channel. Because of the possible difficulties associated with controlling tokens on a decentralized basis (resolving token contention), tokens are expected to be controlled on a centralized basis by the main MCU.

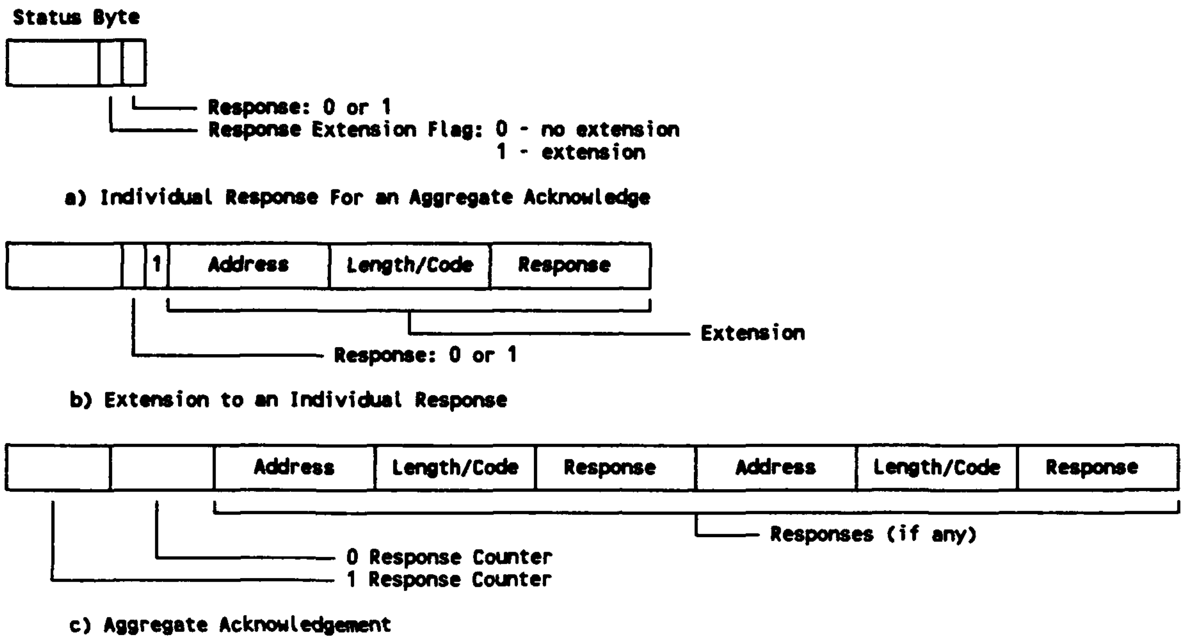

Figure 4-9. Aggregate Acknowledgement Formats

4.2.3 Active Conference Information

Active conference information refers to the type of conference, list of users, capability of users, etc. The Study Group is now discussing what active conference is, who may access it, and whether it should be centralized or distributed.

5.0 MULTIPOINT CONFERENCING CAPABILITIES

The following sections discuss various teleconferencing concerns such as network connectivity, automatic call setup, transfer of control between conferees, multiple function control, and billing.

5.1 Network Connectivity

In general, audiographic conferences rely on a network of audiographic terminals (AGTs) connected together by MCUs in a star pattern. Nevertheless, an "MCU-less" teleconference might be possible if MCU-like capabilities are available to AGTs. For example, on the ISDN, each hookup provides two 64 kb/s B channels. Both channels would be available for use by an AGT, and, theoretically, each B channel could be connected to another AGT. In addition, AGTs could have more than one hookup, and, in essence, would be mini-MCUs. If the CCS and MCS are carefully constructed, these configurations, and others, could be provided without requiring protocol changes from one configuration to another or from one communication network to another.

For example, in the simplest case two AGTs could directly connect to one another (See Figure 5-1). Since each can have MCU-like capabilities, a connection to a separate MCU is unnecessary. This is similar to two Group 3 facsimile machines communicating with one another over the PSTN. Unlike the Group 3 machines, however, additional AGTs can be added, and added without using MCUs. For instance, a third AGT can easily join the original two-AGT conference. By using their unused ISDN B channels, each of the original AGTs has the capability to connect to one more AGT. This type of network can add AGTs ad infinitum: each new AGT becomes the last link in a chain (See Figure 5-2). Unfortunately, this type of network handles leaving AGTs poorly. If an AGT, other than the two end AGTs, decides to leave the conference, the network is broken into two parts; disrupting the conference until the two parts can reestablish communications with one another. The disruption can be minimized if the two AGTs at the ends of the chain connect

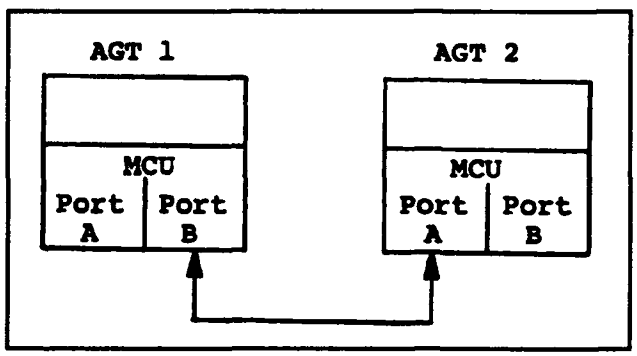

Figure 5-1. AGT to AGT Connection

with one another (See Figure 5-3). Then, if an AGT drops out, all the other AGTs are still connected. Although this connection allows AGTs to leave without disrupting the conference, no new AGTs can easily join the conference (unless the D channel is used to manage connectivity). Future members are potentially locked out.

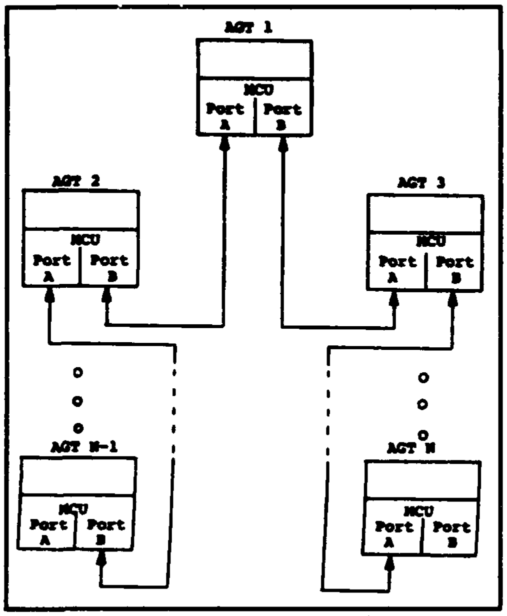

Figure 5-2. Network Consisting of AGTs Only

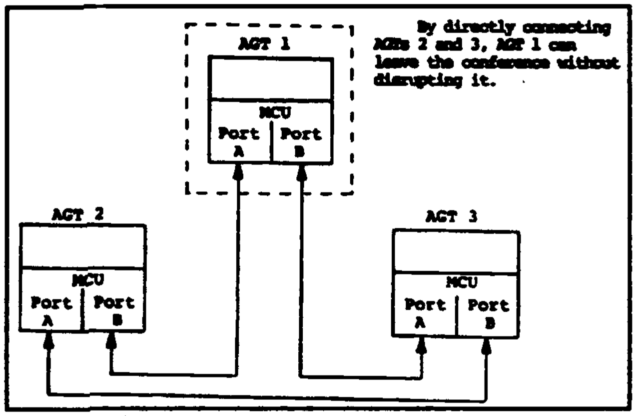

Figure 5-3. Fault Tolerant AGT Network

By adding MCU(s) (or mini-MCUs) to the network, a conference can allow AGTs to join or leave the conference without disruption (See Figure 5-4). Each AGT connects directly to an MCU. If an AGT leaves the conference, it doesn't affect the network or disrupt the conference. In addition, by carefully interconnecting the MCUs and AGTs a fault-tolerant network can be constructed (See Figure 5-5). In the fault-tolerant network, each AGT would connect to two different MCUs, and each MCU would connect to at least two other MCUs. Then, if an MCU fails, its failure would not affect the conference, and the conference could continue without disruption.

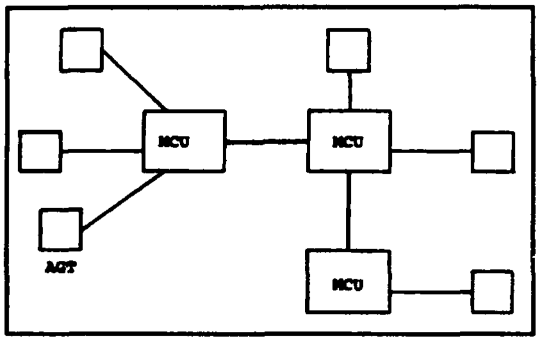

Figure 5-4. Network Containing AGTs and MCUs

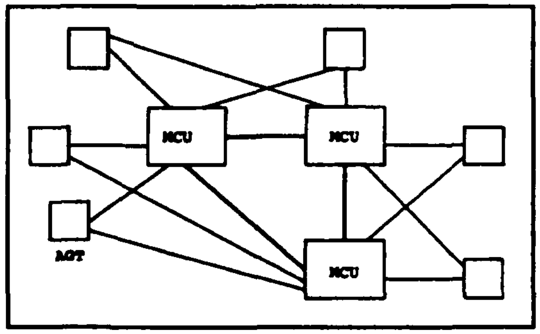

Figure 5-5. Fault Tolerant MCU and AGT Network

Finally, a network can be constructed which is a mixture of the MCU-less network and the MCU network (See Figure 5-6). In it chains of AGTs coexist with AGTs which are directly connected to MCUs. Unfortunately, the AGT chains are still afflicted by the problems associated with AGTs leaving the AGT chains; i.e., their potential for breaking apart the network and disrupting the conference. AGTs can, however, join the conference: they can connect to either an MCU or an AGT.

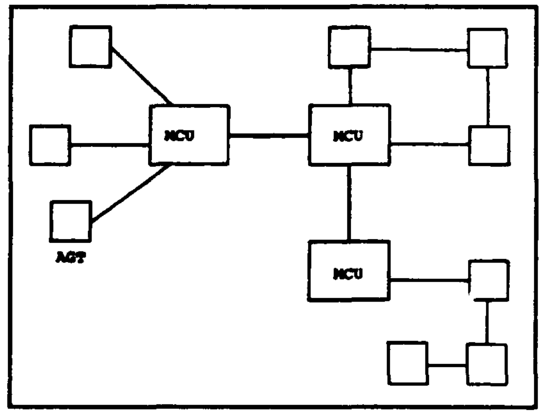

Figure 5-6. Combined MCU and AGT Networks

If a communications network other than ISDN is used (PSDN, PSTN, etc.), each CCS/MCS-based AGT can be viewed as a single port MCU. Thus, two AGTs can still communicate without using a separate MCU. For larger conferences, separate MCUs (or mini-MCUs) are needed to connect all the AGTs.

5.2 Automatic Call Setup

At present, conference sites are not automatically connected to one another. Humans must help establish connections to each site through the communication networks. For example, to establish a teleconference and connect all conference sites, the conference convener sometimes provides a list of all participating sites to a communication network's teleconferencing operators. These operators are then responsible for connecting all the sites just before the conference

begins. It should be possible, by using just a few assumptions, to automate this process:

1. Audiographic terminals and MCUs use the CCS and MCS.
2. Terminals and MCUs can be connected using communication network call numbers.
3. All participating terminal call numbers are known.
4. All participating MCU call numbers are known or are disguised as terminal call numbers.[1]
5. The terminal to MCU call number assignments are also known or can be obtained from the MCUs.[2]

Careful construction of the CCS and MCS, and prior knowledge of terminal and MCU call numbers (or addresses) could make automatic teleconferencing call setup possible. The CCS and MCS would allow different sites with different capabilities to communicate; while prior knowledge of the conference sites' call numbers allow the sites to be automatically connected. If the CCS and MCS are used to perform automatic call setup, and the connection process begins from the convener's terminal, there are at least four cases to consider:

1. The convener has prior knowledge of terminal call numbers, MCU call numbers, and terminal to MCU call number assignments.
2. The convener has prior knowledge of terminal call numbers and MCU call numbers, but terminal to MCU call number assignments are unknown.
3. The convener has prior knowledge of terminal call numbers. The MCU call numbers are disguised as terminal call numbers. Terminal to MCU call number assignments are unknown.
4. The convener has prior knowledge of terminal call numbers, but no MCUs are used. (In this case, terminals are assumed to have "built-in" MCUs capable of handling at least two connections.)

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[1] Disguised MCU call numbers are possible because a company might use its own private MCU to internally connect its own audiographic terminals; while externally it presents its network as a one unified terminal.

[2] Knowing terminal to MCU call number assignments can help reduce billing charges. In general, terminals should be assigned to MCUs with whom their connect charges will be the lowest. For instance, in the U.S., if the PSTN network is used for teleconferencing, the area code and local exchange call numbers could be used to assign the terminals to specific MCUs.

In the first case, where terminal call numbers, MCU call numbers, and terminal to MCU call number assignments are known, the convener's terminal calls its MCU and gives the MCU a list of the conference's MCUs and a list of the number of terminal assignments for each MCU. The convener's MCU uses these lists to define and bootstrap the conference's MCU network. As each MCU is connected, it examines the number of terminal assignments list to assess how many ports it will have free to connect to other MCUs. In general, at least one should be available. Once the number of free ports is determined, an MCU calls at least one other unconnected MCU (See Figure 5-7). As each MCU connects, all connected MCUs decide who will call the next MCU. How this decision is made depends on a number of criteria. For instance, what MCU network configuration will reduce overall connect charges while accounting for the required terminal to MCU assignments. After the MCUs have established their network, they can then get their assigned terminals' call numbers from the convener terminal and call them. Once all terminals are connected, the conference may begin.

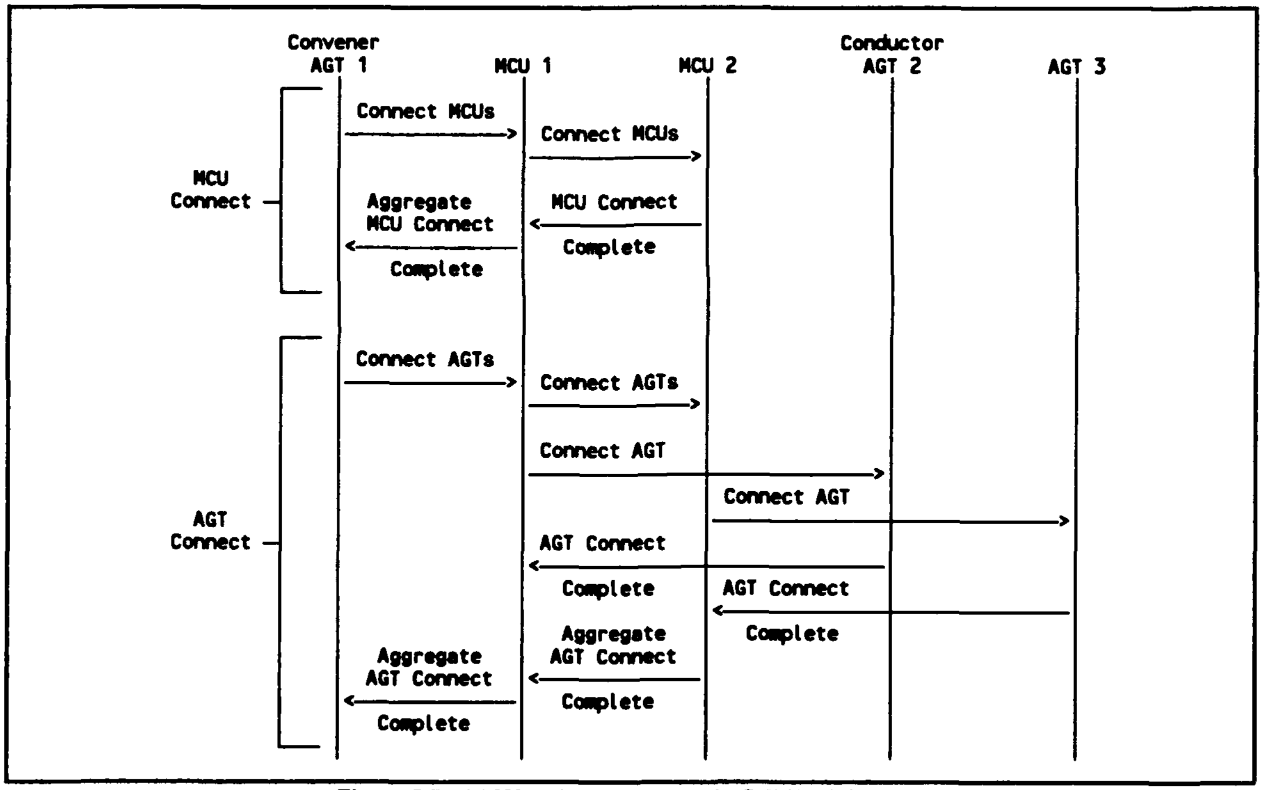

Figure 5-7. MCU and AGT Automatic Call Establishment

In the second case, the terminal to call number assignments are unknown. So, when the convener's terminal calls its MCU, it sends the list of terminal call number instead of the number of terminal assignments for each MCU list. Then as each MCU connects, the MCU

examines the list to determine which terminals can be assigned to it, and how many free ports will remain. After all MCUs are interconnected and each has picked its terminals, any remaining terminals can be assigned.

In the third case, the MCU call numbers are disguised as terminal call numbers. In one approach, the convener's terminal could poll each terminal to determine which are the MCUs: once all are identified, case two would apply.

In the last case, each terminal is assumed to be able to connect to two other terminals. The convener's terminal would call two terminals and give each half of the terminal call numbers. Those terminals would call one terminal on their list, and pass the list to the called terminal. This process would repeat until all terminals are connected (See Figure 5-8).

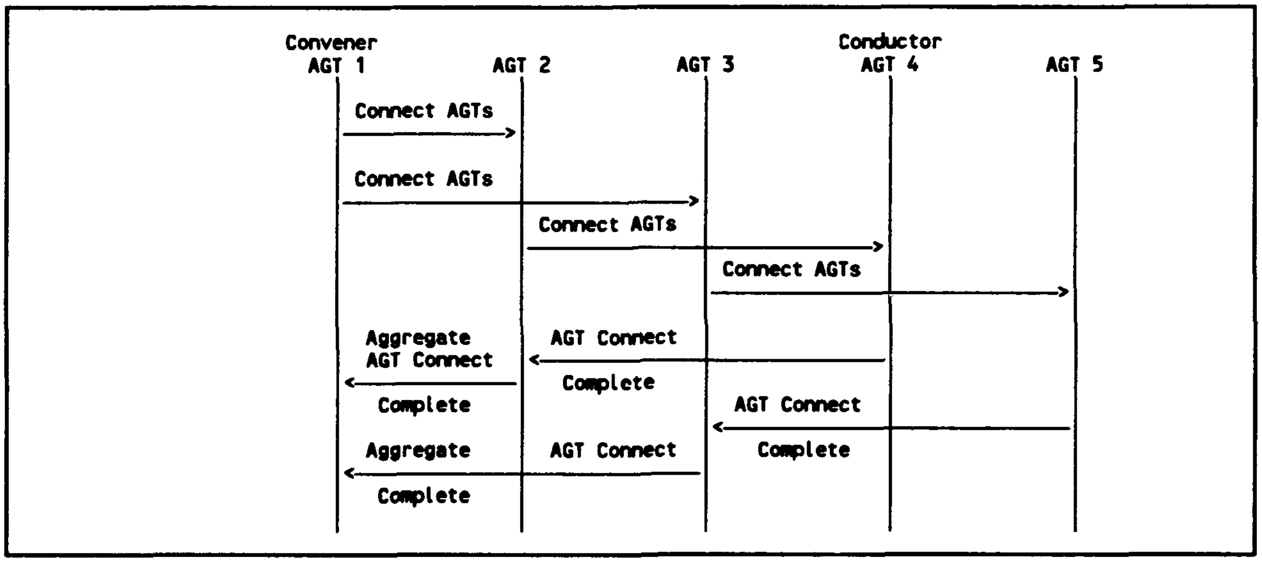

Figure 5-8. AGT Automatic Call Establishment

5.3 Transfer of Control Between Conferees

In a teleconference there are four types of control a participant may have: conference conductor, conference convener, presenter, or audience member. They are ranked according to the following order:

1) Conference conductor (Highest)
2) Conference convener
3) Presenter
4) Audience member (Lowest)

An audience member has no control other than to listen and view what is being presented, or to request the role of presenter. The presenter may use teleconferencing resources to present information to the conference members, e.g. telewriting, facsimile, pointers, etc. Although the presenter may only do so after the conference conductor grants that privilege. The conference convener controls who joins or leaves the conference and initially, during conference setup, assumes the role of conference conductor. The conference conductor controls the conference and determines who the presenter will be, and what resources they may use. Regardless of what the presenter is doing, the conductor always retains the ability to interrupt. In addition, like the convener, the conductor can control who joins or leaves the conference.

In some conferences, sub-conferences might be formed, and each sub-conference, like the main conference, could have a convener, a conductor, a presenter, and audience members. Given the formation of sub-conferences, the question of liaison between sub-conferences, and between the sub-conference and the main conference arises. Thus, in addition to the other four types of conference members, there is a possible fifth type: liaison member. While a sub-conference is in session, the liaison member could be responsible for reporting the proceedings of the sub-conference to other sub-conferences and to the main conference. Likewise, the liaison member could be responsible for collecting information on the proceedings of other sub-conferences and the main conference, and reporting them to his own sub-conference, or more than one liaison member could be appointed. A liaison member could be appointed by a sub-conference for each active sub-conference. A liaison member could have access to the main conference or other sub-conference(s) by either being a member of that conference or by having contact with that conference's liaison personnel.

In the conference rankings, a liaison member could probably be ranked lower than the presenter, but higher than an audience member. In general, a liaison member is just an audience member with an expanded role. The liaison role could probably be held in conjunction with any of the other conference roles: conductor, convener, presenter, etc.

The special role of liaison member raises a few questions:

- Must each sub-conference elect a different liaison member for every other sub-conference plus the main conference, or can one serve all?

- Can liaison members from different sub-conferences form their own sub-conference? (Possibly for the purpose of rapidly updating others on their own sub-conference's progress)
- How many (sub-)conferences can a liaison member be a part of? Need they be?
- Can a liaison member temporarily join another sub-conference? If so, how?
- How do liaison members of different conferences contact one another and exchange information?
- Who appoints liaison members? Can an audience member request the role?

These questions are possible areas of future study.

5.4 Multiple Function Control

Multiple function control refers to the control of all conference resources: facsimile, telewriting, etc. If token control is used, these functions could be assigned to different AGTs. For instance, one AGT might be allowed to send facsimiles to all other conference participants, while the speaker uses telewriting. Thus, all resources needn't be assigned to a single presenter. There can be a facsimile presenter, telewriting presenter, speaker presenter, etc. Thus, while one presenter is speaking and telewriting, another could be transmitting his or her presentation material via facsimile, and thereby maximizing the use of conference resources.

5.5 Billing

Audiographic conferencing uses a variety of services which do not necessarily use the same billing practices. In general, billing charges may be divided into two components: access and utilization.

The access component compensates administrations for the facilities needed by a customer to access a services or services. For the PSTN, a good example is the wiring connecting customers to exchanges.

The utilization component compensates administrations for services provided to the customer. These charges depend upon one or more of the following parameters:

- Service requested
- Service to be provided on demand, reserved, or permanent
- Duration of communication or volume of data transmitted or both
- Distance between correspondents
- Time of use (Allows modulation of charges according to peak or off-peak periods)
- Call set-up
- Call attempts

In general, these charges are established and set on a national level. Given that audiographic conferencing will eventually use the ISDN, the CCITT Recommendations D.200 Series relating to billing charges should probably be used as a basis for determining audiographic services billings.

6.0 RECOMMENDATIONS

A general purpose, multipoint audiographic service might be best served by constructing the CCS and MCS to allow AGTs to have MCU-like capabilities, to be communication network independent, to support tokens, and to support automatic call setup. Tokens permit a very flexible and robust service while allowing for future enhancements. As features are added, new tokens can be added to accommodate them.

Secondly, a modification could be made to the teleconferencing model to allow for sub-conferences and liaison members, the communiques between them, and the communiques between them and the main conference.

7.0 AREAS FOR FUTURE STUDY

The following items could be considered for future study:

- AGTs with MCU-like capabilities
- Connecting non-MCU-like AGTs to MCU-like AGTs
- Token control in non-conducted mode
- Matters of liaison between sub-conferences and the main conference.
  - Must each sub-conference elect a different liaison member for every other sub-conference plus the main conference, or can one serve all?
  - Can liaison members from different sub-conferences form their own sub-conference? (Possibly for the purpose of rapidly updating one another on their sub-conference's progress)
  - How many (sub-)conferences can a liaison member be a part of? Need they be?
  - Can a liaison member temporarily join another sub-conference? If so, how?
  - How do liaison members of different conferences contact one another and exchange information?
  - Who appoints liaison members? Can an audience member request the role?

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